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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/529,280	08/24/2005	Yaakov Stein	RAD 19.779 (100757-00082)	4342

26304 7590 06/25/2008
KATTEN MUCHIN ROSENMAN LLP
575 MADISON AVENUE
NEW YORK, NY 10022-2585

EXAMINER	
LEE, GINA W	

ART UNIT	PAPER NUMBER
2626	

MAIL DATE	DELIVERY MODE
06/25/2008	PAPER

Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)	
	10/529,280	STEIN, YAAKOV	
	Examiner	Art Unit	
	GINA W. LEE	2626	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE *3* MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *23 March 2005*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-23* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-23* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *23 March 2005* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date *23 March 2005*.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

DETAILED ACTION

Specification

1. The specification is objected to as failing to provide proper antecedent basis for the claimed subject matter. See 37 CFR 1.75(d)(1) and MPEP § 608.01(o). Correction of the following is required: the specification does not define the terms "article of manufacture" or "computer usable medium", which are used in claim 19, and it is unclear what they mean. See also objection to claim 19 below.

Claim Objections

2. Claim 19 is objected to because of the following informalities: the phrases "article of manufacture" and "computer usable medium" have not been defined in the specification, and it is unclear what they mean. Although reference is made to a "storage medium" (Specification, page 17, line 16), these two terms are not mentioned. For purposes of examination, the claim and its dependent claim have been treated to exclude computer communications media such as signals or carrier waves. Appropriate correction is required

3.

4. Claim 21 is objected to because of the following informalities: in lines 23-25, the claim states that the "first, second, third, and fourth encoding algorithms are chosen to allow for ... compression of speech, music, video, text, and graphics respectively". However, this states that 4 algorithms correspond to 5 types of data. It is assumed that the word "text" is inserted erroneously, as the body of the claim does not refer to a numbered *encoding algorithm* for text

(step (g) of the claim implies that the text is encoded, but does not make reference to an algorithm). Appropriate correction is required.

Claim Rejections - 35 USC § 102

5. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

6. Claims 1, 5, 6, 10-13, and 16-17 are rejected under 35 U.S.C. 102(e) as being anticipated by Saunders et al. (US 6,351,733), hereinafter referred to as Saunders.

7. With respect to independent **claim 1**, Saunders teaches a system providing low bit-rate compression of data comprising speech and music components for transmission *(col. 2, lines 15-18, audio program is produced so that the audio content is readily fabricated for transmission)*, over a network, said system comprising:

a. a speech encoder encoding said speech component via a first encoding algorithm, transforming said encoded speech signal into a format suitable for transmission *(Fig. 9-12, col. 15, lines 21-24, signal is compressed with a speech-only codec)*, and embedding synchronization information associated with said speech component *(col. 16, lines 15-24; claim 19; audio tracks are time-synchronized and video-frame synchronized to a video signal)*;

b. a music encoder encoding said music component via a second encoding algorithm, said second encoding algorithm different from said first encoding algorithm; transforming said encoded music signal into a format suitable for transmission *(Fig. 9-12, col. 15, lines 21-30; non-speech signal is encoded using a general compression algorithm; col. 16, lines 9-12, two parallel streams are fed into two distinct compression algorithms)*; and embedding synchronization information associated with said music component *(col. 16, lines 15-24; claim 19; audio tracks are time-synchronized and video-frame synchronized to a video signal)*; and

c. a multiplexer multiplexing said outputs of said speech encoder and said music encoder for transmission over said network *(Fig. 12, col. 16, lines 24-28, outputs of compression units are multiplexed in a specific manner so that the audio can be transmitted over a digital medium)*,

wherein said first and second encoding algorithms are chosen to allow for low bit-rate compression of speech and music respectively *(col. 15, lines 27-30, the distinction between compression using the speech-only codec and the general codec helps to reduce the required bandwidth)*.

8. With respect to **claim 5**, Saunders teaches everything claimed, as applied above (see claim 1); in addition, Saunders teaches a system as per claim 1, wherein audio volumes associated with said speech component and said music component are modifiable relative to each other *(col. 6, lines 44-54, the volume of each signal may be independently adjusted; col. 22, lines 53-67; claim 9; the ratio between the levels of the channels may be adjusted)*.

9. With respect to **claim 6**, Saunders teaches everything claimed, as applied above (see claim 1); in addition, Saunders teaches a system as per claim 1, wherein said speech encoder is a LPC, MELP, CELP, or waveform interpolation encoder *(col. 15, lines 24-27, speech coding can be conducted using any known speech codec such as the CELP codec)*.

10. With respect to **claim 10**, Saunders teaches everything claimed, as applied above (see claim 1); in addition, Saunders teaches a system as per claim 1, wherein said music encoder is a transform-based encoder *(col. 20, lines 20-51, audio codecs which may be used include MP3, which is a transform-based encoder)*.

11. With respect to **claim 11**, Saunders teaches everything claimed, as applied above (see claim 1); in addition, Saunders teaches a system as per claim 1, wherein said network is any of the following: local area network, wide area network, the Internet, cellular network, storage network, or wireless network *(col. 9, lines 37-40; col. 25, lines 23-30; transmission may be an ISDN transmission to a PC modem)*.

12. With respect to independent **claim 12**, Saunders teaches a system providing low bit-rate compression of audio comprising speech and music components for transmission over a communication channel *(col. 2,*

lines 15-18, audio program is produced so that the audio content is readily fabricated for transmission), said system comprising:

a. a first analog-to-digital converter converting said speech component into a digital speech signal *(col. 8, lines 16-43; all audio sources utilize microphones to transducer audio information into real-time electrical signals, from which digital masters are created, using a digital format such as PCM)*;

b. a speech encoder encoding said digital speech signal via a first encoding algorithm *(Fig. 9-12, col. 15, lines 21-24, signal is compressed with a speech-only codec)*;

c. a speech audio formatter transforming said encoded speech signal into a format suitable for transmission *(col. 16, lines 24-25, encoded data is ready for multiplexing and transmission)* and embedding synchronization information associated with said speech component *(col. 16, lines 15-24; claim 19; audio tracks are time-synchronized and video-frame synchronized to a video signal)*;

d. a second analog-to-digital converter converting said music component into a digital music signal *(col. 8, lines 16-43; all audio sources utilize microphones to transducer audio information into real-time electrical signals, from which digital masters are created, using a digital format such as PCM)*;

e. a music encoder encoding said digital music signal via a second encoding algorithm, said second encoding algorithm different from said first encoding algorithm *(Fig. 9-12, col. 15, lines 21-*

30; non-speech signal is encoded using a general compression algorithm; col. 16, lines 9-12, two parallel streams are fed into two distinct compression algorithms);

f. a music audio formatter transforming said encoded music signal into a format suitable for transmission *(col. 16, lines 24-25, encoded data is ready for multiplexing and transmission)* and embedding synchronization information associated with said music component *(col. 16, lines 15-24; claim 19; audio tracks are time-synchronized and video-frame synchronized to a video signal)*; and

g. a multiplexer multiplexing said outputs of said speech audio formatter and said music audio formatter for transmission over said channel *(Fig. 12, col. 16, lines 24-28, outputs of compression units are multiplexed in a specific manner so that the audio can be transmitted)*.

13. With respect to **claim 13**, Saunders teaches everything claimed, as applied above (see claim 12); in addition, Saunders further teaches a system as per claim 12, wherein said speech encoder is a LPC, MELP, CELP or waveform interpolation encoder *(col. 15, lines 24-27, speech coding can be conducted using any known speech codec such as the CELP codec)*.

14. With respect to **claim 16**, Saunders teaches everything claimed, as applied above (see claim 12); in addition, Saunders further teaches a system as per claim 12, wherein said music encoder is a transform-based encoder *(col. 20,*

lines 20-51, audio codecs which may be used include MP3, which is a transform-based encoder).

15. With respect to independent **claim 17**, Saunders teaches a method to encode audio for transmission over a communication channel, said audio comprising speech and music components *(col. 2, lines 15-18, audio program is produced so that the audio content is readily fabricated for transmission)*, said method comprising:

a. converting said speech component into a digital speech signal *(col. 8, lines 16-43; all audio sources utilize microphones to transducer audio information into real-time electrical signals, from which digital masters are created, using a digital format such as PCM)*;

b. encoding said digital speech signal via a first encoding algorithm *(Fig. 9-12, col. 15, lines 21-24, signal is compressed with a speech-only codec)*;

c. transforming said encoded speech signal into a format suitable for transmission *(col. 16, lines 24-25, encoded data is ready for multiplexing and transmission)* and embedding synchronization information associated with said speech component *(col. 16, lines 15-24; claim 19; audio tracks are time-synchronized and video-frame synchronized to a video signal)*;

d. converting said music component into a digital music signal *(col. 8, lines 16-43; all audio sources utilize microphones to transducer audio information into real-time electrical signals, from which digital masters are created, using a digital format such as PCM)*;

e. encoding said digital music signal via a second encoding algorithm, said second encoding algorithm different from said first encoding algorithm *(Fig. 9-12, col. 15, lines 21-30; non-speech signal is encoded using a general compression algorithm; col. 16, lines 9-12, two parallel streams are fed into two distinct compression algorithms)*;

f. transforming said encoded music signal into a format suitable for transmission *(col. 16, lines 24-25, encoded data is ready for multiplexing and transmission)* and embedding synchronization information associated with said music component *(col. 16, lines 15-24; claim 19; audio tracks are time-synchronized and video-frame synchronized to a video signal)*; and

g. multiplexing said outputs of steps (c) and (f) for transmission over said channel *(Fig. 12, col. 16, lines 24-28, outputs of compression units are multiplexed in a specific manner so that the audio can be transmitted)*.

Claim Rejections - 35 USC § 103

16. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

17. Claim 19 is rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders et al. (US 6,351,733), hereinafter referred to as Saunders.

18. With respect to independent **claim 19**, Saunders teaches an article of manufacture comprising a computer usable medium having computer

readable program code embodied therein for decoding transmitted data received over a communication channel *(col. 9, lines 37-40; col. 25, lines 23-30; transmission may be an ISDN transmission to a PC modem. While Saunders does not explicitly teach that the method is performed using computer readable program code embodied on a computer usable medium, it at least implies that it is implemented on a computer, since the transmission may be received by a computer)*, said transmitted data comprising a plurality of components, each component encoded via a separate encoding algorithm to provide low bit-rate compression *(col. 16, lines 9-12, two parallel audio streams are fed into two distinct compression algorithms during encoding)*, said medium comprising:

a. computer readable program code aiding in receiving said transmitted data received over said communication channel *(Fig. 13, col. 22, lines 53-54, digital bitstream is received)*;

b. computer readable program code de-multiplexing said data into a plurality of components, said components comprising at least a speech component and a music component *(Fig. 13, col. 22, lines 54-55, two audio signals are separated from each other)*;

c. computer readable program code decoding said speech component via a first decoding algorithm *(Fig. 13, col. 22, lines 54-55, two audio signals are decoded to digital audio signals)*; and

d. computer readable program code decoding said music component via a second decoding algorithm *(Fig. 13, col. 22, lines 54-55, two audio signals are decoded to digital audio signals)*, said second decoding

`algorithm different from said first decoding algorithm` *(see preamble)*.

It is noted that Saunders teaches the encoding process in detail (see claim 1), and that the decoding process is merely the reversal of the same sequence of steps.

19. Claim 2 is rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders et al. (US 6,351,733), hereinafter referred to as Saunders, as applied to claim 1 above, and further in view of Kane et al. (US 5,293,450), hereinafter referred to as Kane.

20. With respect to **claim 2**, Saunders teaches everything claimed, as applied above (see claim 1); but Saunders does not explicitly teach a `system as per claim 1, wherein said data is a composite of said speech and music components and said system further comprises a signal separator, said signal separator separating said speech and music components from said composite.` However, the examiner contends that this concept was well known, as taught by Kane.

In the same field of vocal audio coding, Kane teaches a system to code the voice signal separately from the noise (non-voice) signal *(Fig. 7, circuits 31-33 code the noise)*. Thus a voice may be separated from background music and both coded separately *(col. 5, line 64-col. 6, line 3)*.

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify the system of Saunders with the voice/non-voice separator of Kane, in order to increase the functionality of the system to be able to use audio data that does not already have vocal data separated from other audio data.

21. Claim 7 is rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders et al. (US 6,351,733), hereinafter referred to as Saunders, as applied to claim 1 above, and further in view of Newlin (US 5,774,857).

22. With respect to **claim 7**, Saunders teaches everything claimed, as applied above (see claim 1); but Saunders does not teach a system as per claim 1, wherein said speech encoder is used in conjunction with a speech-to-text converter, and

- said speech-to-text converter converting said speech component to a text component; and
- said speech encoder encoding said text components and formatting said encoded text into a format suitable for transmission.

However, the examiner contends that these concepts were well known in the art, as taught by Newlin.

In the same field of endeavor of encoded audio and video transmission, Newlin teaches a system that accepts the input of an audio signal *(Fig. 3, col. 11, lines 45-48)* and processes the audio signal with a speech recognition subsystem to form a text representation of speech *(Fig. 3, element 307, col. 11, lines 51-54)*. The text is then processed by the closed caption encoder *(Fig. 3, element 311)* to convert it to a closed caption video format *(col. 11, lines 54-61)* to prepare it for mixing with the video signal for transmission *(col. 11, lines 61-67)*.

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify the system of Saunders with the speech-to-text converter of Newlin, in order

to automatically generate a visual representation of speech without use of dedicated systems or user entry of the material to provide access for the hearing impaired *(Newlin, col. 2, line 4-15)*.

23. Claims 8, 15, and 18 are rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders et al. (US 6,351,733), hereinafter referred to as Saunders, in view of Holmes et al. (US 5,506,932), hereinafter referred to as Holmes.

24. With respect to **claim 8**, Saunders teaches everything claimed, as applied above (see claim 1); in addition, Saunders teaches that audio tracks are time-synchronized and video-frame synchronized *(col. 16, lines 15-24; claim 19)* but Saunders does not specifically teach a system as per claim 1, wherein said embedded synchronization information is any of the following: timestamps, synchronization labels, media synchronization tags, synchronizing tokens, or wait-on-event commands. However, the examiner contends that this concept was well known in the art, as taught by Holmes.

In a related field of endeavor of audio and video synchronization, Holmes teaches that audio data is synchronized to the video data on a frame-by-frame basis using a clock *(col. 5, lines 20-23)*. Each frame is identified by a time stamp *(col. 6, lines 23-24)*.

It would have been obvious to one of ordinary skill in the art at the time the invention was made to implement the synchronization taught by Saunders with the method using a time stamp as taught by Holmes, because the time stamp method is one of a finite number of methods known to be useful for audio-video data stream synchronization, as a person with ordinary skill has good reason to pursue the known options within his or her technical grasp. Using the known

method would have had predictable results, so it would have been obvious to try the time stamp synchronization method to synchronize the data.

25. With respect to **claim 15**, Saunders teaches everything claimed, as applied above (see claim 12); in addition, Saunders teaches that audio tracks are time-synchronized and video-frame synchronized *(col. 16, lines 15-24; claim 19)* but Saunders does not specifically teach a system as per claim 12, wherein said embedded synchronization information is any of the following: timestamps, synchronization labels, media synchronization tags, synchronizing tokens, or wait-on-event commands. However, the examiner contends that this concept was well known in the art, as taught by Holmes.

In a related field of endeavor of audio and video synchronization, Holmes teaches that audio data is synchronized to the video data on a frame-by-frame basis using a clock *(col. 5, lines 20-23)*. Each frame is identified by a time stamp *(col. 6, lines 23-24)*.

It would have been obvious to one of ordinary skill in the art at the time the invention was made to implement the synchronization taught by Saunders with the method using a time stamp as taught by Holmes, because the time stamp method is one of a finite number of methods known to be useful for audio-video data stream synchronization, as a person with ordinary skill has good reason to pursue the known options within his or her technical grasp. Using the known method would have had predictable results, so it would have been obvious to try the time stamp synchronization method to synchronize the data.

26. With respect to **claim 18**, Saunders teaches everything claimed, as applied above (see claim 17); in addition, Saunders teaches that audio tracks are time-synchronized and video-frame synchronized *(col. 16, lines 15-24; claim 19)* but Saunders does not specifically teach a method as per claim 17, wherein said embedded synchronization information is any of the following: timestamps, synchronization labels, media synchronization tags, synchronizing tokens, or wait-on-event commands. However, the examiner contends that this concept was well known in the art, as taught by Holmes.

In a related field of endeavor of audio and video synchronization, Holmes teaches that audio data is synchronized to the video data on a frame-by-frame basis using a clock *(col. 5, lines 20-23)*. Each frame is identified by a time stamp *(col. 6, lines 23-24)*.

It would have been obvious to one of ordinary skill in the art at the time the invention was made to implement the synchronization taught by Saunders with the method using a time stamp as taught by Holmes, because the time stamp method is one of a finite number of methods known to be useful for audio-video data stream synchronization, as a person with ordinary skill has good reason to pursue the known options within his or her technical grasp. Using the known method would have had predictable results, so it would have been obvious to try the time stamp synchronization method to synchronize the data.

27. Claims 9 and 14 are rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders et al. (US 6,351,733), hereinafter referred to as Saunders, in view of Rabowsky et al. (US 5,680,512).

28. With respect to **claim 9**, Saunders teaches everything claimed, as applied above (see claim 1); but Saunders does not specifically teach a `system as per claim 1, wherein said music encoder is a MIDI-encoder or linear musical score notation.` However, the examiner contends that this concept was well known in the art, as taught by Rabowsky.

In the same field of endeavor of audio coding, Rabowsky teaches separate encoding of a voice and non-voice signal, which are combined to form the audio signal *(col.1, lines 49-60)*. The encoder for the non-voice signal *(Fig. 1, element 12)* may use MIDI *(col. 5, lines 19-25, signal is applied to the encoder; col. 5, lines 43-57, MIDI elements are stored for reference in the encoder)*.

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify the audio coding system of Saunders with the MIDI encoder of Rabowsky, in order to increase the functionality of the coder to encode MIDI data.

29. With respect to **claim 14**, Saunders teaches everything claimed, as applied above (see claim 12); but Saunders does not specifically teach a `system as per claim 12, wherein said music encoder is a MDI-encoder or linear musical score notation.` However, the examiner contends that this concept was well known in the art, as taught by Rabowsky.

In the same field of endeavor of audio coding, Rabowsky teaches separate encoding of a voice and non-voice signal, which are combined to form the audio signal *(col.1, lines 49-60)*. The encoder for the non-voice signal *(Fig. 1, element 12)* may use MIDI *(col. 5, lines 19-25,*

signal is applied to the encoder; col. 5, lines 43-57, MIDI elements are stored for reference in the encoder).

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify the audio coding system of Saunders with the MIDI encoder of Rabowsky, in order to increase the functionality of the coder to encode MIDI data.

30. Claims 3, 4, and 20-23 are rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders et al. (US 6,351,733), hereinafter referred to as Saunders, in view of Tsukagoshi (US 6,104,861), and further in view of Ito et al. (US 2001/0012444), hereinafter referred to as Ito.

31. With respect to **claim 3**, Saunders teaches everything claimed, as applied above (see claim 1); in addition, Saunders teaches that the audio may be synchronized with associated video *(col. 10, lines 29-31; col. 16, lines 21-24)* such as a motion picture, but Saunders does not explicitly teach a system as per claim 1, wherein said data further comprises a text component, a video component, and a graphics component, said system further comprising:

- a text formatter transforming said text component into a format suitable for transmission and embedding synchronization information associated with said text component;
- a video encoder encoding said video component via a third encoding algorithm, said third encoding algorithm different from said first and second encoding algorithms; transforming said encoded video signal into a format suitable for

transmission; and embedding synchronization information associated with said video component;

- a graphics encoder encoding said graphics component via a fourth encoding algorithm, said fourth encoding algorithm different from said first, second, and third encoding algorithms; transforming said encoded graphics into a format suitable for transmission; and embedding synchronization information associated with said graphics component; and
- said multiplexer in (c) additionally multiplexing the output of said text formatter, said video encoder, and graphics encoder.

However, the examiner contends that these concepts were well known in the art, as taught by Tsukagoshi and Ito.

In the same field of endeavor of encoding and decoding of multiple types of data streams, Tsukagoshi teaches a system for encoding a video broadcast into multiple types of data streams for transmission. The system includes: a subtitle encoder *(Fig. 9A, element 57)* that generates subtitles *(col. 13, lines 16-29)*, selects data for compression and encodes it *(col. 13, lines 30-55)*, and forwards the subtitle information to the multiplexer for multiplexing with the audio and video data to be transmitted *(col. 14, lines 33-44)*; a video encoder *(Fig. 9A, element 52)* which compresses digital video for video transmission *(col. 13, lines 1-8)* and synchronized with the subtitle data *(col. 13, lines 5-8)* and multiplexed with the other data by the multiplexer *(col. 14, lines 33-44)*; as well as an audio encoder *(Fig. 9A, lines 9-15)*.

It would have been obvious to one of ordinary skill in the art at the time the invention was made to use the audio encoding method of Saunders in the data encoding method of

Tsukagoshi because Tsukagoshi teaches an efficient way of transmitting video broadcast comprised of multiple data streams, as suggested by Saunders *(col. 10, lines 29-31; col. 16, lines 21-24).*

In the same field of endeavor of encoding and decoding of multiple types of data streams, Ito teaches a system for encoding a broadcast for transmission. The system comprises separate speech encoder *(Fig. 18, element 5001)* and a character object encoder *(Fig. 18, element 5004)*, as well as a synthesized image object encoder *(Fig. 18, element 5003, paragraphs [0087-0088]).* The output of the encoder is multiplexed with the outputs of the other encoders and output as a bitstream for transmission *(Fig. 18).* A synthesized image may be a computer graphic *(paragraph [0088])* such as a background object or weather information image *(paragraph [0257]).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to further modify the system of Saunders and Tsukagoshi with the additional graphic encoder of Ito, because the addition of graphics to the video would enable more information to be represented graphically to viewers, especially in such applications as a news program *(paragraph [0257]).*

32. With respect to **claim 4**, Saunders in view of Tsukagoshi and Ito teach everything claimed, as applied above (see claim 3); in addition, Saunders does not but Tsukagoshi does teach a system as per claim 3, wherein said text component corresponds to subtitles associated with said video components *(Fig. 9A, col. 13, lines 16-29, information from character generator that is encoded is subtitles)*.

It would have been obvious to one of ordinary skill in the art at the time the invention was made to use the audio encoding method of Saunders in the data encoding method of Tsukagoshi because Tsukagoshi teaches an efficient way of transmitting video broadcast comprised of multiple data streams, as suggested by Saunders *(Saunders, col. 10, lines 29-31; col. 16, lines 21-24).*

33. With respect to **claim 20**, Saunders teaches everything claimed, as applied above (see claim 19); in addition, Saunders teaches that the audio may be synchronized with associated video *(col. 10, lines 29-31; col. 16, lines 21-24)* such as a motion picture. However, while Saunders does not, Tsukagoshi teaches an article of manufacture as per claim 19, wherein said plurality of components additionally comprises a video component, a text component, and a graphics component, said medium further comprising:

a. in addition to de-multiplexing said data into speech and music component, computer readable program code de-multiplexing said video component, said text component, and said graphics component

b. computer readable program code formatting said text component;

c. computer readable program code decoding said video component via a third decoding algorithm, said third decoding algorithm different from said first and second decoding algorithm; and

d. computer readable program code decoding said graphics component via a fourth decoding algorithm, said fourth decoding algorithm different from said first, second, and third decoding algorithm.

However, the examiner contends that these concepts were well known in the art, as taught by Tsukagoshi and Ito.

In the same field of endeavor of encoding and decoding of multiple types of data streams, Tsukagoshi teaches a system for decoding a video broadcast transmitted in multiple types of data streams. The system includes a demultiplexer to multiplex the signal into audio, video, and subtitle streams *(Fig. 1, element 1, col. 4, lines 22-28)*. The subtitle data is decoded and mixed with the decoded video data *(Fig. 1, col. 4, lines 52-56)* and the video data is decoded and converted for display *(Fig. 1, col. 4, lines 31-51)*. Each of the audio, video, and subtitle streams are decoded separately *(Fig. 1)*.

It would have been obvious to one of ordinary skill in the art at the time the invention was made to use the audio decoding method of Saunders in the data decoding method of Tsukagoshi because Tsukagoshi teaches an efficient way of transmitting and decoding video broadcast comprised of multiple data streams, as suggested by Saunders *(col. 10, lines 29-31;*

In the same field of endeavor of encoding and decoding of multiple types of data streams, Ito teaches decoding a transmitted broadcast. After the system has received and demultiplexed the data *(Fig. 1, element 36)*, image data is decoded by the image decoding circuit (72) having a plurality of decoding units to decode respective image objects *(paragraph [0266])*. A type of image object is a synthesized image *(paragraph [0084])*, which may be a computer graphic

(paragraph [0088]) such as a background object or weather information image *(paragraph [0257]).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to further modify Saunders and Tsukagoshi with the additional graphic decoder of Ito, because the addition of graphics to the video would enable more information to be represented graphically to viewers, especially in such applications as a news program *(paragraph [0257]).*

It is noted that Saunders in view of Tsukagoshi and Ito teaches the corresponding encoding method (see claim 3) and the decoding process is the reversal of the same sequence of steps.

34. With respect to independent **claim 21**, Saunders teaches `a method encoding data for transmission over a communication network, said data comprising speech, music ... components` *(col. 2, lines 15-18, audio program is produced so that the audio content is readily fabricated for transmission)*, `said method comprising the steps of:`

`a. encoding said speech component via a first encoding algorithm` *(Fig. 9-12, col. 15, lines 21-24, signal is compressed with a speech-only codec)*;

`b. transforming said encoded speech signal into a format suitable for transmission` *(col. 16, lines 24-25, encoded data is ready for multiplexing and transmission)* and `embedding synchronization information associated with said speech component` *(col. 16, lines 15-24; claim 19; audio tracks are time-synchronized and video-frame synchronized to a video signal)*;

c. encoding said music component via a second encoding algorithm, said second encoding algorithm different from said first encoding algorithm *(Fig. 9-12, col. 15, lines 21-30; non-speech signal is encoded using a general compression algorithm; col. 16, lines 9-12, two parallel streams are fed into two distinct compression algorithms)*;

d. transforming said encoded music signal into a format suitable for transmission *(col. 16, lines 24-25, encoded data is ready for multiplexing and transmission)*; and embedding synchronization information associated with said music component *(col. 16, lines 15-24; claim 19; audio tracks are time-synchronized and video-frame synchronized to a video signal)*;

wherein said first, second … encoding algorithms are chosen to allow for low bit-rate compression of speech, music … respectively *(col. 15, lines 27-30, the distinction between compression using the speech-only codec and the general codec helps to reduce the required bandwidth)*.

Saunders also teaches that the compressed audio outputs are multiplexed so that the audio can be transmitted *(Fig. 12, col. 16, lines 24-28*; but Saunders does not teach

e. encoding said video component via a third encoding algorithm, said third encoding algorithm different from said first and second encoding algorithms;

f. transforming said encoded video signal into a format suitable for transmission and embedding synchronization information associated with said video component;

g. transforming a text component into a format suitable for transmission and embedding synchronization information associated with said text component;

j. multiplexing said outputs of steps … (f) [and] (g)… for transmission over said network,

wherein said … third … encoding algorithms are chosen to allow for low bit-rate compression of … video [and] text… respectively.

However, the examiner contends that these concepts were well known in the art, as taught by Tsukagoshi.

In the same field of endeavor of encoding and decoding of multiple types of data streams, Tsukagoshi teaches a system for encoding a video broadcast into multiple types of data streams for transmission. The system includes: a subtitle encoder *(Fig. 9A, element 57)* that generates subtitles *(col. 13, lines 16-29)*, selects data for compression and encodes it *(col. 13, lines 30-55)*, and forwards the subtitle information to the multiplexer for multiplexing with the audio and video data to be transmitted *(col. 14, lines 33-44)*; a video encoder *(Fig. 9A, element 52)* which compresses digital video for video transmission *(col. 13, lines 1-8)* and synchronized with the subtitle data *(col. 13, lines 5-8)* and multiplexed with the other data by the multiplexer *(col. 14, lines 33-44)*; as well as an audio encoder *(Fig. 9A, lines 9-15)*.

It would have been obvious to one of ordinary skill in the art at the time the invention was made to use the audio encoding method of Saunders in the data encoding method of Tsukagoshi because Tsukagoshi teaches an efficient way of transmitting video broadcast

comprised of multiple data streams, as suggested by Saunders *(col. 10, lines 29-31; col. 16, lines 21-24).*

The combination of Saunders and Tsukagoshi does not teach:

h. encoding said graphics component via a fourth encoding algorithm, said fourth encoding algorithm different from said first, second, and third encoding algorithm;

i. transforming said encoded graphics signal into a format suitable for transmission; and embedding synchronization information associated with said graphics component; and

j. multiplexing said outputs of steps … (i) for transmission over said network,

wherein said … fourth encoding algorithms are chosen to allow for low bit-rate compression of … graphics respectively.

However, the examiner contends that these concepts were well known in the art, as taught by Ito.

In the same field of endeavor of encoding and decoding of multiple types of data streams, Ito teaches a system for encoding a broadcast for transmission. The system comprises separate speech encoder *(Fig. 18, element 5001)* and a character object encoder *(Fig. 18, element 5004)*, as well as a synthesized image object encoder *(Fig. 18, element 5003, paragraphs [0087-0088]).* The output of the encoder is multiplexed with the outputs of the other encoders and output as a bitstream for transmission *(Fig. 18).* A synthesized image may be a computer graphic *(paragraph [0088])* such as a background object or weather information image *(paragraph [0257]).*

It would have been obvious to one of ordinary skill in the art at the time the invention was made to further modify the system of Saunders and Tsukagoshi with the additional graphic encoder of Ito, because the addition of graphics to the video would enable more information to be represented graphically to viewers, especially in such applications as a news program *(paragraph [0257])*.

35. With respect to **claim 23**, Saunders teaches everything claimed, as applied above (see claim 21); in addition, Saunders further teaches a method as per claim 21, wherein said network is any of the following: local area network, wide area network, the Internet, cellular network, storage area network, or wireless network *(col. 9, lines 37-40; col. 25, lines 23-30; transmission may be an ISDN transmission to a PC modem)*.

36. Claim 22 is rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders et al. (US 6,351,733), hereinafter referred to as Saunders, in view of Tsukagoshi (US 6,104,861) and Ito et al. (US 2001/0012444), hereinafter referred to as Ito, as applied to claim 21 above, and further in view of Holmes et al. (US 5,506,932), hereinafter referred to as Holmes.

37. With respect to **claim 22**, Saunders in view of Tsukagoshi and Ito teaches everything claimed, as applied above (see claim 21); in addition, Saunders teaches that audio tracks are time-synchronized and video-frame synchronized *(col. 16, lines 15-24; claim 19)* but Saunders does not specifically teach a method as per claim 21, wherein said embedded synchronization information is any of the following: timestamps, synchronization labels, media synchronization tags,

`synchronizing tokens, or wait-on-event commands.` However, the examiner contends that this concept was well known in the art, as taught by Holmes.

In a related field of endeavor of audio and video synchronization, Holmes teaches that audio data is synchronized to the video data on a frame-by-frame basis using a clock *(col. 5, lines 20-23)*. Each frame is identified by a time stamp *(col. 6, lines 23-24)*.

It would have been obvious to one of ordinary skill in the art at the time the invention was made to implement the synchronization taught by Saunders with the method using a time stamp as taught by Holmes, because the time stamp method is one of a finite number of methods known to be useful for audio-video data stream synchronization, as a person with ordinary skill has good reason to pursue the known options within his or her technical grasp. Using the known method would have had predictable results, so it would have been obvious to try the time stamp synchronization method to synchronize the data.

Conclusion

38. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

Imai et al. (JP 05-056007), Vaudrey et al. (US 6,442,278; US 6,985,594; US 6,311,155), Saunders et al. (US 2002/0040295), and Marko et al. (US 7,075,946) teach methods of encoding, transmitting, and decoding audio signals with separate speech and background audio signals.

Sie et al. (US 5,534,941), Sherman et al. (US 6,535,269), Tsukagoshi (US 5,748,256) teach systems with multiple data streams.

Mead (US 6,088,484) and Fallon (US 6,624,761) teach using different methods of encoding for different types of data.

39. Any inquiry concerning this communication or earlier communications from the examiner should be directed to GINA W. LEE whose telephone number is (571)270-3139. The examiner can normally be reached on Monday to Friday, 8:00 AM - 5:00 PM EST.

40. If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Patrick Edouard can be reached on (571) 272-7603. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

41. Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

Patrick Edouard
Examiner
Art Unit 2626

GWL

/Patrick N. Edouard/
Supervisory Patent Examiner, Art Unit 2626